User Manual

BudgeTone - 200 Series

IP Phone

For Firmware Version 1.1.0.16

Grandstream Networks, Inc.

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GRANDSTREAM NETWORKS
Innovative IP Telephony

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1 Welcome

Thank you for purchasing Grandstream BudgeTone-200 IP Phone. You made an excellent choice and we hope you will enjoy all its capabilities.

Grandstream's BudgeTone-200 SIP IP phone is the innovative IP telephone that offers a rich set of functionality and superb sound quality. They are fully compatible with SIP industry standard and can interoperate with many other SIP compliant devices and software on the market.

This document is subject to changes without notice. The latest electronic version of this user manual is available for download from the following location:

http://www.grandstream.com/user_manuals/BudgeTone200.pdf

2 Installation

2.1 What is Included in the Package

The BudgeTone-200 phone package contains:

1. One BudgeTone -200 Main Case
2. One Handset
3. One Phone Cord
4. One Universal Power Adapter
5. One Ethernet Cable

2.2 Connecting Your Phone

Following is a backside picture of BudgeTone-200, each connection port is labeled with the name in the following table:-

The table below describes the connectors on the BudgeTone-200 phone:

LAN	10/100 Switch LAN port for connecting to Ethernet.
PC	10/100 Switch port for connecting PC
POWER	5V power port
HEADSET	2.5mm Headset port

2.3 Safety Compliances

The BudgeTone-200 phone is compliant with various safety standards including FCC/CE. Its power adaptor is compliant with UL standard. The phone should only be operated with the universal power adaptor provided with the package. Damages to the phone caused by using other unsupported power adaptors are not covered by the manufacturer's warranty.

2.4 Warranty

Grandstream has a reseller agreement with our reseller customer. End user should contact the company from whom you purchased the product for replacement, repair or refund.

If you purchased the product directly from Grandstream, contact your Grandstream Sales and Service Representative for a RMA (Return Materials Authorization) number.

Grandstream reserves the right to remedy warranty policy without prior notification.

Warning: Please do not attempt to use a different power adaptor. Using other power adaptor may damage the BudgeTone-200 and will void the manufacturer warranty.

Caution: Changes or modifications to this product not expressly approved by Grandstream, or operation of this product in any way other than as detailed by this User Manual, could void your manufacturer warranty.

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3 Product Overview

The following photo illustrates the appearance of a BudgeTone-200 IP phone.

Front View

Side View

3.1 Key Features

Grandstream BudgeTone-200 IP Phone is a next generation IP telephone based on industry open standard SIP (Session Initiation Protocol). Built on innovative technology, Grandstream IP Phone features market leading superb sound quality and rich functionalities at mass-affordable price.

Software Features:

- Support SIP 2.0, TCP/UDP/IP, PPPoE, RTP/RTCP, HTTP, ARP/RARP, ICMP, DNS, DHCP, NTP/SNTP, TFTP.
- Support multiparty conferencing
- Supports Quick IP Call Mode.
- Support NAT traversal using IETF STUN and Symmetric RTP
- Advanced Digital Signal Processing (DSP) technology to ensure superior hi-fidelity audio quality, interoperable with various 3rd party SIP end user device, Proxy/Registrar/Server and Gateway products
- Advanced and patent pending adaptive jitter buffer control, packet delay and loss concealment technology
- Support popular codecs including G711 (a-law and u-law), G.723.1 (6.3K), G.729A/B and GSM. Dynamic negotiation of codec and voice payload length
- Support standard voice features such as Caller ID Display or Block, Call Waiting, Call Waiting Caller ID, Call Hold, Call Transfer (attended/blind), Do-Not-Disturb, Call Forwarding, in-band and out-of-band DTMF(RFC2833), SIP INFO, Dial Plans, Off-Hook Auto Dial, Auto Answer, Early Dial and Speed Dial, etc.
- Full duplex hands-free speakerphone, redial, call log, volume control, voice mail with indicator, downloadable ring tone, etc.
- Support Silence Suppression, VAD (Voice Activity Detection), CNG (Comfort Noise Generation), Line Echo Cancellation (G.168) and AGC (Automatic Gain Control)
- Support Acoustic Echo Cancellation (AEC) with Acoustic Gain Control (AGC) for speakerphone mode
- Support sidetone
- Support DIGEST authentication and encryption using MD5 and MD5-sess
- Provide easy configuration through manual operation (phone keypad), Web interface or automated provisioning by downloading encrypted configuration file via HTTP/TFTP for mass deployment
- Support for Layer 2 (802.1Q VLAN, 802.1p) and Layer 3 QoS (ToS, DiffServ, MPLS)
- Support firmware upgrade via TFTP or HTTP.
- Support DNS SRV Look up and SIP Server Fail Over
- Acoustic Echo Cancellation (AEC) with Acoustic Gain Control (AGC) for speakerphone mode
- Support for Authenticating configuration file before accepting changes

- allow user to specify different URL for configuration file and firmware files

Hardware Features:

- Support Headset which will auto switch to Headset when plugged in
- Support 10/100 Full/Half Duplex Ethernet Switch with LAN and PC port, Ethernet polarity can be auto detected, thus either straight through or twist cable can be used.
- Support Message Waiting Indication LED

3.2 Hardware Specification

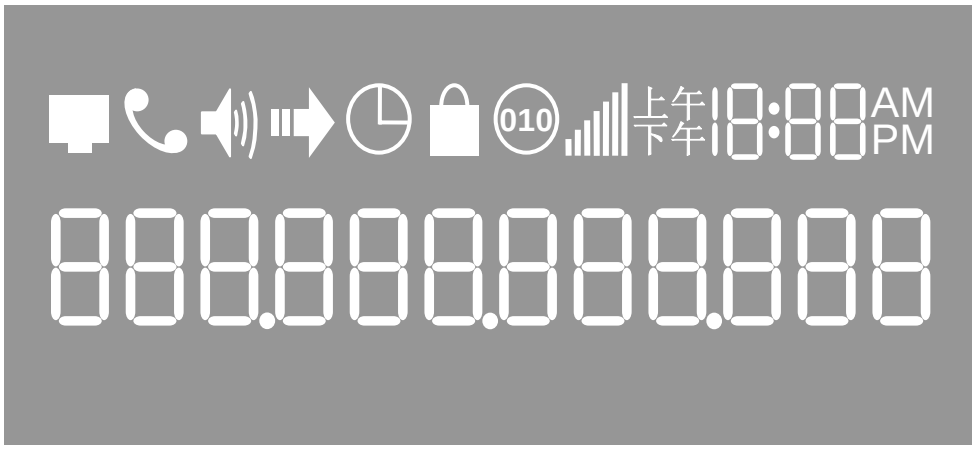
The table below describes the hardware specification of BudgeTone-200:

Model	**BudgeTone-200**
LAN interface	2xRJ45 10/100Base-T
Headset Jack	2.5mm Headset port
LED	1 LED in RED color
Phone Case	25-button keypad 12-digit caller ID LCD
Universal Switching Power Adaptor	Input: 100-240VAC 50-60 Hz Output: +5VDC, 1200mA, **UL** certified
Dimension	18cm (W) 22cm (D) 6.5cm (H)
Weight	0.9kg (2lbs)
Temperature	40 - 130^{o}F 5 – 45^{o}C
Humidity	10% - 90% (non-condensing)
Compliance	FCC / CE / C-Tick

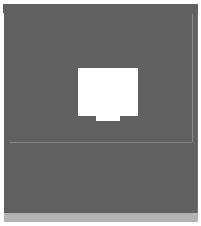
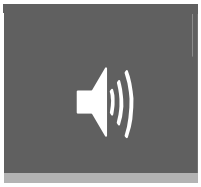
4 Using BudgeTone-200 IP Phone

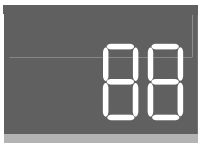
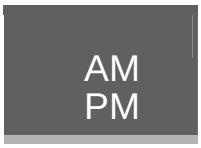
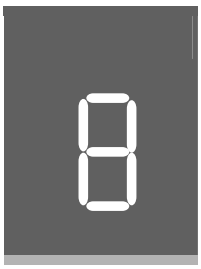
4.1 Getting Familiar with LCD / LED

BudgeTone-200 phone has a numeric LCD of 64mmx24mm size with backlight. This model has a small red LED status reminder. Here is the display when all segments illuminate:

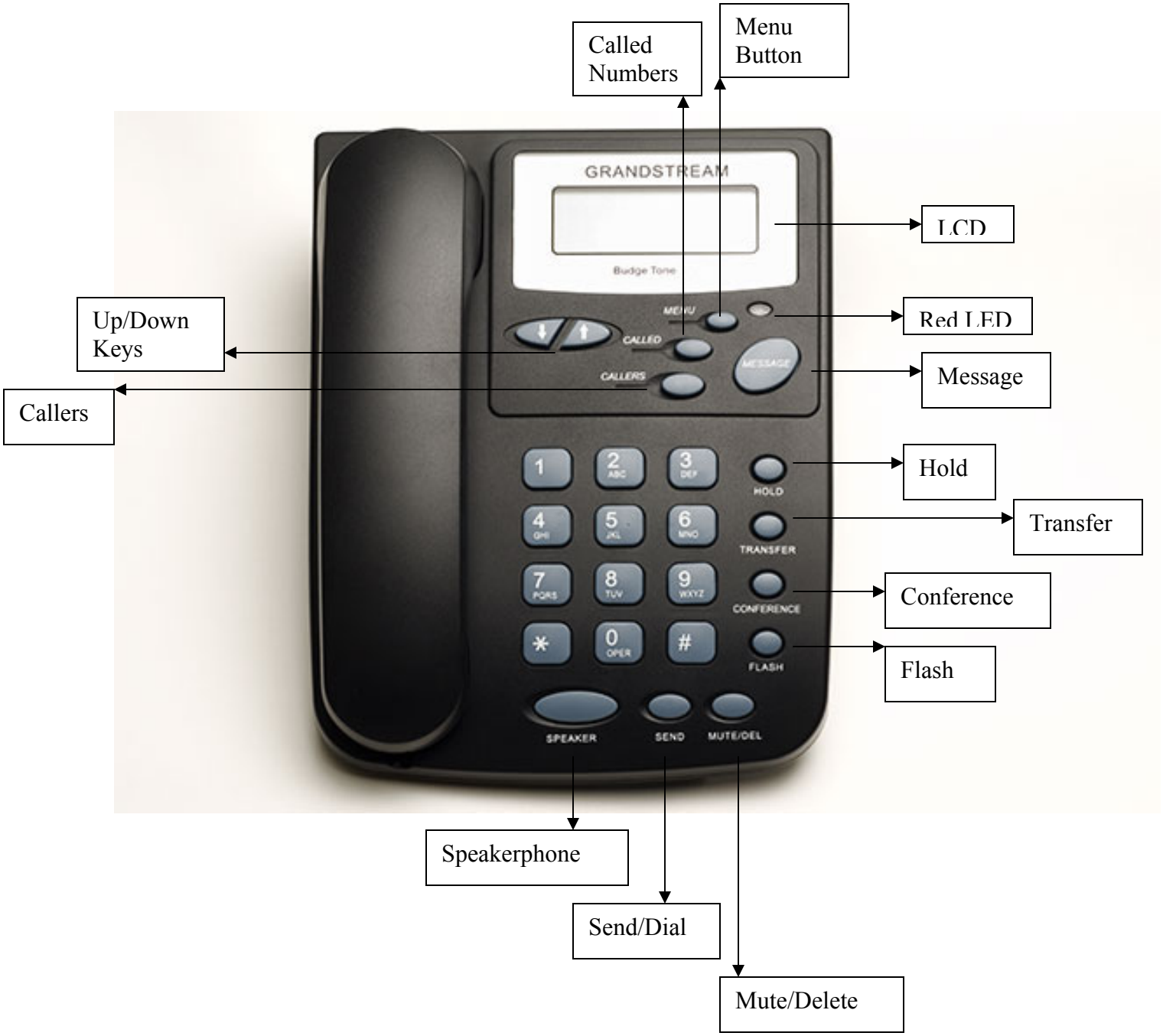


When the phone is in the normal idle state, the backlight is off. Whenever an event (call) occurs, the backlight will turn on automatically to bring the user's attention. In addition, if Voice Mail configured and there is a VM waiting, the red LED will be blinking to remind user there is a Voice Mail in the Voice Mail server.

Icon	LCD Icon Definitions
	Network Status Icon: FLASH in the case of Ethernet link failure or the phone is not registered properly. OFF if IP address or SIP server is not found ON if IP address and SIP server are located
	Phone Status Icon: OFF when the handset is on-hook ON when the handset is off-hook
	Speakerphone/Headset Status Icon: FLASH when phone rings OFF when the speakerphone/headset is off ON when the speakerphone/headset is on

	Handset and Speakerphone/Headset Volume Icons: 0-7 scales to adjust handset / speakerphone volume
	Real-time Clock: Synchronized to Internet time server Time zone configurable via web browser
	Call Logs: 01-10 for *CALLED* history (dialed number) 01-10 for *CALLERS* history (Incoming caller ID)
	Time Icon: AM for the morning PM for the afternoon
	IP Address Separator Icons:
	Numerical Numbers and Characters: 0 - 9 * = ⌞ # = ⌟ A, b, C, c, d, E, F, G, g, H, h, I, L, n, O, o, P, q, r, S, t, U, u, Y

4.2 Getting Familiar with Keypad



Key Button	Key Button Definitions
0 - 9, *, #	Digit, star and pound keys are usually used to make phone calls
↓	1) Reduce handset, speakerphone/headset volume after off hook the phone via handset or speaker 2) Reduce ring tone volume when phone in IDLE and off hook to confirm the changed ring tone volume 3) Next menu item browsing when phone is in IDLE mode after MENU key pressed, off hook to interrupt and exit
↑	1) Increase handset, speakerphone/headset volume after off hook the phone via handset or speaker 2) Increase ring tone volume when phone in IDLE and off hook to confirm the changed ring tone volume 3) Previous menu item browsing when phone is in IDLE mode after MENU key pressed, off hook to interrupt and exit
MENU	Enter keypad MENU mode when phone is in IDLE mode. It is also the ENTER key once entering MENU
CALLED	After off hook, press to display the dialed numbers. When number displayed, press the SEND key can make call using that displayed number
CALLERS	After off hook, press to display the incoming Caller IDs. When number displayed, press the SEND key can make call using that displayed number
MESSAGE	Enter to retrieve voice mails from Voice Mail Portal or Server
HOLD	Temporarily hold the active call
TRANSFER	Transfer the active call to another party
CONFERENCE	Establish 3-way conferencing call
FLASH	Flash event to switch between two lines
MUTE/DEL	Mute an active call; or Delete a key entry, call log etc Also used to 'REJECT' incoming call.

SEND/(RE)DIAL	Dial a new number inputted or Redial the number last dialed. After entering the phone number, pressing this key would force a call to go out immediately before timeout
SPEAKERPHONE	Enter hands-free mode

4.3 Making and Answering Phone Calls

4.3.1 Handset, Speakerphone and Headset Mode

The regular Handset mode can be switched with either the Speaker mode (Hand free) or the Headset mode, however, whenever the Headset is plugged in, Speaker mode will be switched to the Headset mode automatically.

Handset mode and Speakerphone/Headset mode cannot be enabled at the same time. Pressing the hook-switch or Speakerphone button would toggle the phone between these two modes.

To Switch between Handset and Speaker/Headset, simply press the Hook Flash in the Handset cradle or the Speaker button.

4.3.2 Make Calls using Numbers

There are FIVE ways to make phone calls:

- Pick up handset or press SPEAKERPHONE button, and then enter the phone numbers
- Press the SEND button directly to redial the number last called.
- Once pressed, the last dialed number will be displayed on the LCD as the corresponding DTMF tones are played out and an outgoing call is sent.
- Browse the CALLED/CALLER history and press the SEND/REDIAL button.
- Pick up the handset or press the speakerphone button, then press the “CALLED/CALLERS” button to browse thru the last 10 numbers dialed out. Once the desired number is identified and displayed on the LCD screen, press the SEND button and a new call to that displayed number will be sent out immediately.

Examples:

- To dial another extension on the same proxy, such as 1008, simply pick up handset or press speakerphone, dial 1008 and then press the “SEND” button.
- To dial a PSTN number such as 6266667890, you might need to enter in some prefix number followed by the phone number. Please check with your VoIP service provider to get the information. If you phone is assigned with a PSTN-like number such as 6265556789, most likely you just follow the rule to dial 16266667890 as if you were calling from a regular analog phone, followed by pressing the “SEND” button.

4.3.3 Make Calls using IP Address

Direct IP calling allows two parties, that is, a BudgeTone phone and another VoIP Device, to talk to each other in an ad hoc fashion without a SIP proxy. This kind of VoIP calls can be made between two parties if:

- Both BudgeTone phone and other VoIP Device(i.e., another IP Phone or BudgeTone SIP phone or other VoIP unit) have public IP addresses, or
- Both BudgeTone phone and other VoIP Device are on the same LAN using private or public IP addresses, or
- Both BudgeTone phone and other VoIP Device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

This model has the ability to dial an IP address under the same LAN segment by simply pressing the last octet in the IP address.

In the Advanced Settings page there is an option "Use Quick IP-call mode", by default it is set to No. When this option is set to YES, and #XXX is dialed, where X is 0-9 and XXX <=255, phone will make direct IP call to aaa.bbb.ccc.XXX where aaa.bbb.ccc comes from the local IP address REGARDLESS of subnet mask.

#XX or #X are also valid so leading 0 is not required (but OK).

eg.
192.168.0.2 calling 192.168.0.3 just dial #3 follow by SEND or #
192.168.0.2 calling 192.168.0.23 just dial #23 follow by SEND or #
192.168.0.2 calling 192.168.0.123 just dial #123 follow by SEND or #

192.168.0.2 dial #3 and #03 and #003 has same effect --> call 192.168.0.3

Note:- If you have a SIP Server configured, Direct IP-IP call will still work. However, if you are using STUN, Direct IP-IP call will also use STUN.

OR

To make a direct IP to IP call, first off hook, then press *"MENU"* key, then enter a 12-digit target IP address to make the call. If port is not default 5060, destination ports can be specified by using "*4" (encoding for ":") followed by the port number.

Examples:

- If the target IP address is 192.168.0.10, the dialing convention is
  MENU_key 192 168 000 010
  followed by pressing the "SEND" key or wait for seconds in the No Key Entry Timeout.

- If the target IP address/port is 192.168.1.20:5062, then the dialing convention would be:
  MENU_key 192168001020*45062
  followed by pressing the "SEND" key wait for seconds in the No Key Entry Timeout.

4.3.4 Answer an Incoming Call

There are two ways to answer an incoming call:

1. Pick up the handset to answer the call normally using handset, or
2. Press the SPEAKERPHONE button to answer in speakerphone or headset mode

4.3.5 Call Hold

While in conversation, pressing the "Hold" button will put the remote end on hold. Pressing the "Hold" button again will release the previously Hold state and resume the bi-directional media.

4.3.6 Call Waiting and Call Flashing

If call waiting feature is enabled, while the user is in a conversation, he will hear a special stutter tone if there is another incoming call. User then can press FLASH button to put the current call party on hold automatically and switch to the other call. Pressing flash button toggles between two active calls.

4.3.7 Call Transfer

Two transfer operations are supported.

4.3.7.1 Blind Transfer

User can transfer an active call to a third party without announcement.

User presses the "TRANSFER" button and if the other voice channel is available (i.e., there is no other active conversation besides the current one), user will hear a dial tone. User can then dial the third party's phone number followed by pressing SEND button.

NOTE:

- "*Enable Call Feature" has to be configured to "Yes" in web configuration page in order to make the features to work.*

A can hold on to the phone and wait for one of the three following behaviors:

- A quick confirmation tone (temporarily using the call waiting indication tone) follows by a dial tone. This indicates the transfer has been successful. At this point, the user can either hang up or make another call.
- A quick busy tone followed by a restored call (On supported platforms only). This means the transfer has failed due to the failed response sent from server and the phone will try to recover the call. The busy tone is just to indicate to the transferor that the transfer has failed.
- Busy tone keeps playing. This means the phone has failed to receive the final response and decide to time out. Be advised that this does not indicate the transfer has been successful, nor does it indicate the transfer has failed.

4.3.7.2 Attended Transfer

User can transfer an active call to a third party with announcement.

User presses the "FLASH" button and hears a dial tone, then dial the third party's phone number followed by pressing "SEND" button. If the call is answered, press "TRANSFER" to complete the transfer operation and hand up, if the call is not answered, pressing "FLASH" button to resume the original call.

NOTE:

- *When Attended Transfer failed, if A hangs up, the BudgeTone phone will ring user A back again to remind A that B is still on the call. A can pick up the phone to restore conversation with B.*

4.3.8 Conference Call

BudgeTone 200 phone supports 3-way conference.

Assuming that call party A and B are in conversation. A wants to bring C in a conference:

1. A presses the "CONFERENCE" button to get a dial tone and put B on hold
2. A dials C's number then "SEND" key to make the call
3. If C answers the call, then A presses "CONFERENCE" button to bring B, C in the conference.
4. If C does not answer the call, A can press FLASH back to talk to B.

NOTE:

- *During the conference, if B or C drops the call, the remaining two parties can still talk. However, if A the conference initiator hangs up, all calls will be terminated.*

4.3.9 Checking Message and Message Waiting Indication

When BudgeTone-200 is on-hook, pressing the MESSAGE button will trigger the phone to call the VM Server (VMS) configured for the Account.

The MWI (Message Waiting Indicator) LED will flash in red color in three quarters of a second when voicemail server sends message waiting information to BudgeTone-200.

4.3.10 Mute and Delete

When in conversation with an ACTIVE LINE, pressing "MUTE/DEL" will mute the conversation, that is, you can hear the other party but the other party cannot hear you. Pressing the button again will resume the conversation.

When dialing a number, press "MUTE/DEL" will delete the last entered digit.

When receiving incoming call, press "MUTE/DEL" will Reject the call and forward to voice mail.

4.4 Call Features

BudgeTone-200 series phone supports a list of call features: Caller ID Block (or Anonymous Call), Disable/Enable Call Waiting, Call Forward on Busy, Delay, or Unconditional, etc.

Following table shows the call features of BudgeTone-200 series phone.

Key	Call Features
*30	Block Caller ID (for all subsequent calls)
*31	Send Caller ID (for all subsequent calls)
*67	Block Caller ID (per call)
*82	Send Caller ID (per call)
*70	Disable Call Waiting. (Per Call)
*71	Enable Call Waiting (Per Call)
*72	Unconditional Call Forward To use this feature, dial "*72" and get the dial tone. Dial the forward number and "#" for a dial tone, then hang up.
*73	Cancel Unconditional Call Forward To cancel "Unconditional Call Forward", dial "*73" and get the dial tone, then hang up.
*90	Busy Call Forward To use this feature, dial "*90" and get the dial tone. Dial the forward number and "#" for a dial tone, then hang up.
*91	Cancel Busy Call Forward To cancel "Busy Call Forward", dial "*91" and get the dial tone, then hang up.
*92	Delayed Call Forward To use this feature, dial "*92" and get the dial tone. Dial the forward number and "#" for a dial tone, then hang up.
*93	Cancel Delayed Call Forward To cancel this Forward, dial "*93" and get the dial tone, then hang up.

5 Configuration Guide

5.1 Configuration with Keypad

When the phone is IDLE or On Hook, press the *MENU* button to enter key pad menu state. When the phone goes off-hook or a call comes in, the phone automatically exits the key pad menu state and prepare for the call. It also exits the key pad menu state if left idle for 20 seconds.

Here are the key pad menu options supported:

Menu Item	Menu Functions
1	Display "**[1] dhcP On** " or "**[1] dhcP oFF**" for the current selection Press *MENU* key to enter edit mode Press '↓' or '↑' to toggle the selection Press *MENU* to save and exit Must recycle power to take effective!!!
2	Display "**[2] IP Addr** " Press *MENU* to display the current IP address Enter new IP address if DHCP is OFF Press '↓' or '↑' to exit Press *MENU* to (save and) exit Must recycle power to take effective!!!
3	Display "**[3] SubNet** " Press *MENU* to display the Subnet mask Enter new Subnet mask if DHCP is OFF Press '↓' or '↑' to exit Press *MENU* to (save and) exit Must recycle power to take effective!!!
4	Display "**[4] routEr** " Press *MENU* to display the Router/Gateway address Enter new Router/Gateway address if DHCP is OFF Press '↓' or '↑' to exit Press *MENU* to (save and) exit Must recycle power to take effective!!!

Menu Item	Menu Functions
5	Display "**[5] dnS**" Press *MENU* to display the DNS address Enter new DNS address if DHCP is OFF Press '↓' or '↑' to exit Press *MENU* to (save and) exit Must recycle power to take effective!!!
6	Display "**[6] tFtP**" Press *MENU* to display the TFTP address Enter new TFTP server IP address Press *MENU* to save Press '↓' or '↑' to exit
7	Display "**[7] G-711u 2**" Press *MENU* to select new codec Press '↓' or '↑' to browse a list of available codecs line 2 "**- G-711A 2**" 3 "**- G-723 1**" 4 "**- G-729 2**" 5 "**- GS 1**" Press 1 to 9 to indicate number of frames per TX packet Press *MENU* to save and exit Must recycle power to take effective!!!
8	Display "**[8] SIP SP-1**" Reserve for future products.
9	Display "**[9] codE rEL**" Press *Menu* to display the code releases Press '↓' or '↑' to browse line 1 "**b 2006-03-14**" – date: *boot* code 2 "**1. 1. 0. 1**" – version: *boot* code 3 "**P 2006-04-28**" – date: *phone* code 4 "**1. 1. 0. 16** – version: *phone* code 5 "1**r 2004-05-12**" – date: 1st *ring* tone 6 "**0. 0. 0. 0**" – version: *ring* tone 7 "2**r 2004-05-12**" – date: 2nd *ring* tone 8 "**0. 0. 0. 0**" – version: *ring* tone 9 "3**r 0000-00-00**" – date: 3rd *ring* tone 10 "**0. 0. 0. 0**" – version: *ring* tone (all zeroes means unavailable or unsupported) Press *MENU* to exit

Menu Item	Menu Functions
10	Display "[**10] Phy Addr**" Press *MENU* to display the physical / MAC address Press '↓' or '↑' to exit
11	Display "[11] **ring 0**" Press *MENU* to hear the selected ring tone, press '↓' or '↑' to select the stored ring tones. Now only 3 are available, ring 0 (default), ring 1 and ring 2. ring 3 is unavailable or unsupported. Press *MENU* to select and exit
	Display " -- rESEt --", please be very **CAREFUL** here • Key in the physical / MAC address on back of the phone, Press MENU, phone will be reset to FACTORY DEFAULT setting, and all your setting will be erased. • Press MENU key without key in anything, phone will function the same as power cycle or reboot
Others	When phone is powered on and time is displayed • Press '↓' or '↑', Display " ring [4] ", press '↓' or '↑' again to hear and adjust the ring tone volume, from 0 (off) to 7 (maximum), off and on hook to set • Press "SPEAKERPHONE" button, or off hook and pick up handset, press '↓' or '↑' to adjust the speakerphone/headset or handset volume

5.2 Configuration with Web Browser

BudgeTone 200 series IP phone has an embedded Web server that will respond to HTTP GET/POST requests. It also has embedded HTML pages that allow a user to configure the IP phone through a Web browser such as Microsoft's IE.

5.2.1 Access the Web Configuration Menu

The IP Phone Web Configuration Menu can be accessed by the following URI:

http://Phone-IP-Address

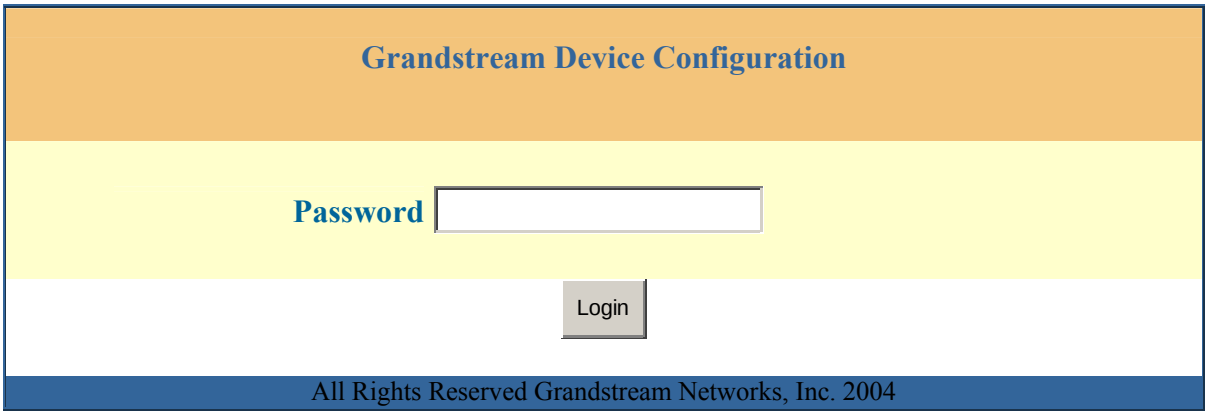
where the *Phone-IP-Address* is the IP address of the phone.
When the phone is *on-hook*, press *Menu* button and then select the Status item to see "**IP: IP Address**"

NOTE:

- *To type IP address into browser to get into the configuration page, please strip out the leading "0" as the browser will parse in octet. e.g.: if the IP address is: 192.168.001.014, please type in: 192.168.1.14.*

5.2.2 End User Configuration

Once this HTTP request is entered and sent from a Web browser, the BudgeTone 200 will respond with the following login screen:



The password is case sensitive with maximum length of 25 characters and the factory default password for End User is “123”.

After a correct password is entered in the login screen, the embedded Web server inside the BudgeTone 200 will respond with the Configuration page which is explained in details below.

Grandstream Device Configuration

STATUS | BASIC SETTINGS | ADVANCED SETTINGS | ACCOUNT

End User Password: (purposely not displayed for security protection)

IP Address:

dynamically assigned via DHCP (default) or PPPoE
(will attempt PPPoE if DHCP fails and following is non-blank)

PPPoE account ID:	
PPPoE password:	
Host name (Option 12):	
Domain name (Option 15):	
Vendor Class ID (Option 60):	

Preferred DNS server: . . .

statically configured as:

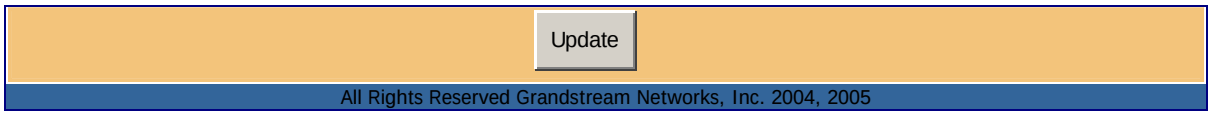
IP Address:	192 . 168 . 0 . 160
Subnet Mask:	0 . 0 . 0 . 0
Default Router:	0 . 0 . 0 . 0
DNS Server 1:	0 . 0 . 0 . 0
DNS Server 2:	0 . 0 . 0 . 0

Time Zone: GMT-7:00 (US Mountain Time, Denver)

Allow DHCP Option 2 to override Time Zone setting:
No Yes

Daylight No Yes (if set to Yes, display time will be 1 hour ahead of

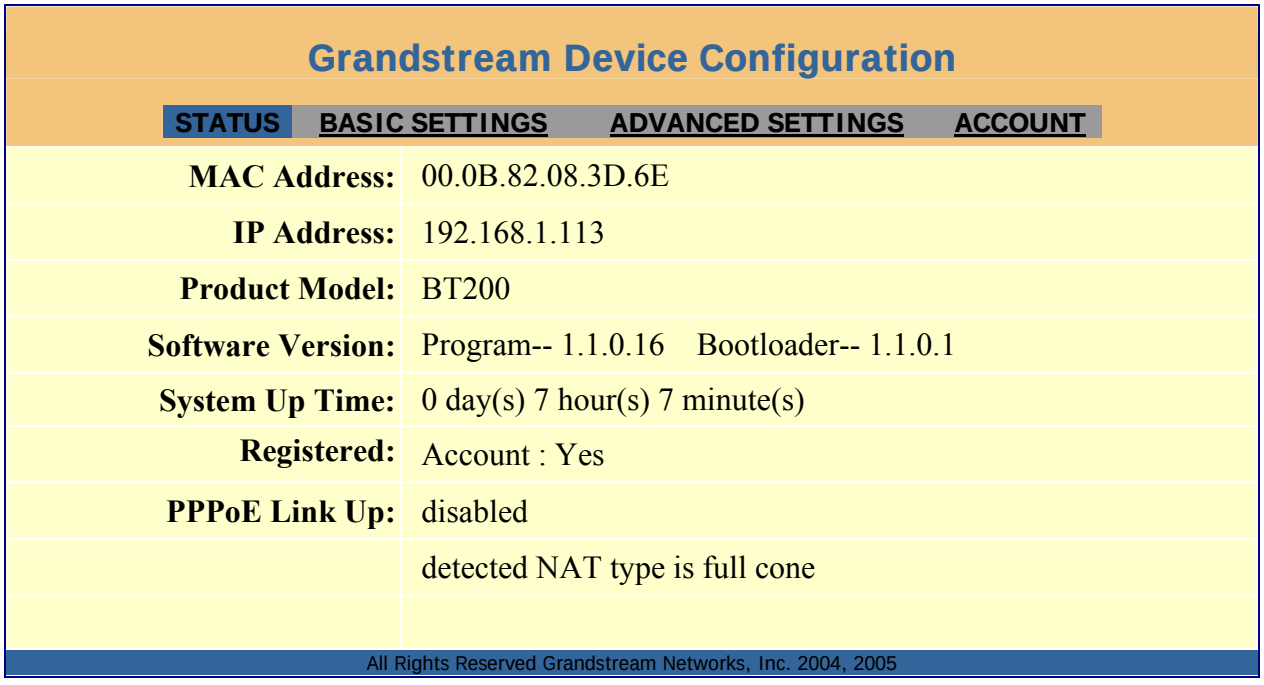
Savings Time:	normal time)
Date Display Format:	Year-Month-Day Month-Day-Year Day-Month-Year
System Device Mode	
Device Mode:	Switch (default) NAT/Router
NAT/Router Configuration	
WAN side http access:	No Yes (WAN side access to http server will be rejected if set to No)
Reply to ICMP on WAN port:	No Yes (Unit will not respond to PING from WAN side if set to No)
Cloned WAN MAC Addr:	(in hex format)
LAN Subnet Mask:	255.255.255.0 (default is 255.255.255.0)
LAN DHCP Base IP:	192.168.2.1 (base IP for the LAN port, default is 192.168.2.1)
DHCP IP Lease Time:	120 (in units of hours, default is 120 hours or 5 days)
DMZ IP:	
Port Forwarding:	WAN port LAN IP LAN port Protocol 0 0 UDP Only 0 0 UDP Only 0 0 UDP Only 0 0 UDP Only 0 0 UDP Only 0 0 UDP Only 0 0 UDP Only 0 0 UDP Only



End User Password	This contains the password to access the Web Configuration Menu. This field is case sensitive with a maximum length of 25 characters.
IP Address	There are two modes under which the BudgeTone 200 can operate: • If DHCP mode is enabled, then all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory.) The BudgeTone 200 will acquire its IP address from the first DHCP server it discovers from the LAN it is connected. • To use the PPPoE feature the PPPoE account settings need to be set. The BudgeTone 200 will attempt to establish a PPPoE session if any of the PPPoE fields is set. • If Static IP mode is enabled, then the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary) fields will need to be configured. These fields are set to zero by default.
Time Zone	This parameter controls how the date/time is displayed according to the specified time zone.
Daylight Savings Time	This parameter controls whether the time will be displayed in daylight savings time or not. If set to "Yes", then the displayed time will be 1 hour ahead of normal time.
Date Display Format	Allow user to choose among the following three formats: Year-Month-Day Month-Day-Year Day-Month-Year
Device Mode	This parameter controls whether the device is working in NAT router mode or Bridge mode. Need save the setting and reboot the device before the setting start to work.
WAN side http access	If set to "Yes", user can access the configuration page through the WAN port, instead of connecting PC and BT200 through the "PC" port to do the configuration. On the other hand, it exposes the BT200 to others, and may cause some security issues for users. Default is No.
Reply to ICMP on WAN port	If set to "Yes", The BT200 will respond to the PING command from other computers for testing, but it also is vulnerable to the DOS attack. Default is No.

Cloned WAN MAC Addr	Allow the user to set a specific MAC address. Set in Hex format.
LAN Subnet Mask	Sets the LAN subnet mask. Default value is 255.255.255.0
LAN DHCP Base IP	Base IP for the LAN port, which function as a Gateway for the subnet. Default value is 192.168.2.1.
DHCP IP Lease Time	Value is set in units of hours. Default value is 120hr (5 Days.) The time IP address is assigned to the LAN clients.
DMZ IP	Forward all WAN IP traffic to a specific IP address if no matching port is used by HandyTone-486 itself or in the defined port forwarding.
Port Forwarding	Allow the user to forward a matching (TCP/UDP) port to a specific LAN IP address with a specific (TCP/UDP) port.

In addition to the Basic Settings configuration page, end user also has access to the device Status page. The following is a screen shot of the device Status page. Details are explained next.



MAC Address	The device ID, in HEX format. This is a very important ID for ISP troubleshooting.
IP Address	This field shows LAN IP address of BudgeTone 200
Product Model	This field contains the product model info.
Software Version	• Program: This is the main software release, its number is always used for firmware upgrade. • Bootloader: This is normally not changed.
System Up Time	This field shows system up time since the last reboot.
Registered	This field indicates whether the device is registered to the SIP server(s).
PPPoE Link Up	This field shows whether the PPPoE connection is up if connected to DSL modem.
Detected NAT Type	This field shows what kind NAT the BudgeTone 200 is connected to via its LAN port. It is based on STUN protocol.

5.2.3 Advanced User Configuration

To login to the Advanced User Configuration page, please follow the instructions in section 5.2.1 to get to the following login page. The password is case sensitive with a maximum length of 25 characters and the factory default password for Advanced User is "admin".

Advanced User configuration includes not only the end user configuration, but also advanced configuration such as SIP configuration, Codec selection, NAT Traversal Setting and other miscellaneous configuration. Following is a screen shot of the advanced configuration page:

Grandstream Device Configuration

STATUS | BASIC SETTINGS | ADVANCED SETTINGS | ACCOUNT

Admin Password:	(purposely not displayed for security protection)
Silence Suppression:	No Yes
Voice Frames per TX:	2 (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)
Layer 3 QoS:	48 (Diff-Serv or Precedence value)
Layer 2 QoS:	802.1Q/VLAN Tag 0 802.1p priority value 0 (0-7)
No Key Entry Timeout:	4 (in seconds, default is 4 seconds)
Use # as Dial Key:	No Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)
local RTP port:	5004 (1024-65535, default 5004)
Use random port:	No Yes
keep-alive interval:	20 (in seconds, default 20 seconds)
Use NAT IP	(if specified, this will be used in SIP/SDP message)
STUN server:	(URI or IP:port)
Firmware Upgrade and Provisioning:	Upgrade Via TFTP HTTP Firmware Server Path: fm.grandstream.com/gs Config Server Path: fm.grandstream.com/gs Firmware File Prefix: Firmware File Postfix: Config File Prefix: Config File Postfix:

	Allow DHCP Option 66 to override server: ○ No ○ Yes Automatic Upgrade: ○ No ○ Yes, check for upgrade every 10080 minutes (default 7 days) ○ Always Check for New Firmware ○ Check New Firmware only when F/W pre/suffix changes
Authenticate Conf File:	○ No ○ Yes (cfg file would be authenticated before acceptance if set to Yes)
DTMF Payload Type:	101
Syslog Server:	
Syslog Level:	NONE
NTP Server:	time.nist.gov (URI or IP address) Allow DHCP Option 42 to override NTP server: ○ No ○ Yes
Distinctive Ring Tone:	Custom ring tone 1, used if incoming caller ID is Custom ring tone 2, used if incoming caller ID is Custom ring tone 3, used if incoming caller ID is
Disable Call-Waiting:	○ No ○ Yes
Use Quick IP-call mode:	○ No ○ Yes
Lock keypad update:	○ No ○ Yes (configuration update via keypad is disabled if set to Yes)

Update

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Admin Password	Administrator password. Only administrator can configure the "Advanced Settings" page. Password field is purposely left blank for security reason after clicking update and saved. The maximum password length is 25 characters.
Silence Suppression	This controls the silence suppression/VAD feature of G723 and G729. If set to "Yes", when a silence is detected, small quantity of VAD packets (instead of audio packets) will be sent during the period of no talking. If set to "No", this feature is disabled.
Voice Frames per TX	This field contains the number of voice frames to be transmitted in a single packet. When setting this value, the user should be aware of the requested packet time (used in SDP message) as a result of configuring this parameter. This parameter is associated with the first vocoder in the above vocoder Preference List or the actual used payload type negotiated between the 2 conversation parties at run time. e.g., if the first vocoder is configured as G723 and the "Voice Frames per TX" is set to be 2, then the "ptime" value in the SDP message of an INVITE request will be 60ms because each G723 voice frame contains 30ms of audio. Similarly, if this field is set to be 2 and if the first vocoder chosen is G729 or G711 or G726, then the "ptime" value in the SDP message of an INVITE request will be 20ms. If the configured voice frames per TX exceeds the maximum allowed value, the BudgeTone 200 will use and save the maximum allowed value for the corresponding first vocoder choice. The maximum value for PCM is 10(x10ms) frames; for G726, it is 20 (x10ms) frames; for G723, it is 32 (x30ms) frames; for G729/G728, 64 (x10ms) and 64 (x2.5ms) frames respectively.
Layer 3 QoS	This field defines the layer 3 QoS parameter which can be the value used for IP Precedence or Diff-Serv or MPLS. Default value is 48.
Layer 2 QoS	This contains the value used for layer 2 VLAN tag. Default setting is blank.
No Key Entry Timeout	Default is 4 seconds.
Use # as Send Key	This parameter allows users to configure the "#" key to be used as the "Send" (or "Dial") key. If set to "Yes", pressing this key will immediately trigger the sending of dialed string collected so far. In this case, this key is essentially equivalent to the "(Re)Dial" key. If set to "No", this "#" key will then be included as part of the dial string to be sent out.

Local RTP port	This parameter defines the local RTP-RTCP port pair the BudgeTone 200 will listen and transmit. It is the base RTP port for channel 0. When configured, channel 0 will use this port _value for RTP and the port_value+1 for its RTCP; channel 1 will use port_value+2 for RTP and port_value+3 for its RTCP. The default value is 5004.
Use Random Port	This parameter, when set to Yes, will force random generation of both the local SIP and RTP ports. This is usually necessary when multiple BudgeTone 200s are behind the same NAT.
Keep-alive interval	This parameter specifies how often the BudgeTone 200 sends a blank UDP packet to the SIP server in order to keep the "hole" on the NAT open. Default is 20 seconds.
Use NAT IP	NAT IP address used in SIP/SDP message. Default is blank.
STUN Server	IP address or Domain name of the STUN server.
Firmware Upgrade and provisioning	This radio button will enable BudgeTone 200 to download firmware or configuration file through either TFTP or HTTP.
Via TFTP Server	This is the IP address of the configured TFTP server. If selected and it is non-zero or not blank, the BudgeTone 200 will attempt to retrieve new configuration file or new code image from the specified TFTP server at boot time. It will make up to 3 attempts before timeout and then it will start the boot process using the existing code image in the Flash memory. If a TFTP server is configured and a new code image is retrieved, the new downloaded image will be verified and then saved into the Flash memory. Note: Please do NOT interrupt the TFTP upgrade process (especially the power supply) as this will damage the device. Depending on the network environment this process can take up to 15 or 20 minutes.
Via HTTP Server	The URL for the HTTP server used for firmware upgrade and configuration via HTTP. For example, http://provisioning.mycompany.com:6688/Grandstream/1.0.5.16 Here ":6688" is the specific TCP port that the HTTP server is listening at, it can be omitted if using default port 80. Note: If Auto Upgrade is set to No, BudgeTone 200 will only do HTTP download once at boot up.
Allow DHCP Option 66 to override server	DHCP Option 66 is used to identify a TFTP server when the 'sname' field in the DHCP header has been used for DHCP options. If you choose yes, BT200 will use the TFTP server resolved from DHCP, instead of the one you specified in the "TFTP Server" option above.

Automatic Upgrade	Choose Yes to enable automatic upgrade and provisioning. In "Check for new firmware every" field, enter the number of days to enable BudgeTone 200 to check the server for firmware upgrade or configuration in the defined period of days. When set to No, BudgeTone 200 will only do upgrade once at boot up. "Always check for New Firmware" "Check New Firmware only when F/W pre/suffix changes"
Authenticate Conf File	if set to Yes, cfg file would be authenticated before acceptance. This mechanism is useful for the protection of configuration on the device from unauthorized change.
DTMF Payload Type	This parameter sets the payload type for DTMF using RFC2833.
Syslog Server	The IP address or URL of System log server. This feature is especially useful for ITSP (Internet Telephone Service Provider)
Syslog Level	Select the ATA to report the log level. Default is NONE. The level is one of DEBUG, INFO, WARNING or ERROR. Syslog messages are sent based on the following events: • product model/version on boot up (INFO level) • NAT related info (INFO level) • sent or received SIP message (DEBUG level) • SIP message summary (INFO level) • inbound and outbound calls (INFO level) • registration status change (INFO level) • negotiated codec (INFO level) • Ethernet link up (INFO level) • SLIC chip exception (WARNING and ERROR levels) • memory exception (ERROR level) The Syslog uses USER facility. In addition to standard Syslog payload, it contains the following components: GS_LOG: [device MAC address][error code] error message Here is an example: May 19 02:40:38 192.168.1.14 GS_LOG: [00:0b:82:00:a1:be][000] Ethernet link is up
NTP server	URI or IP address of the NTP (Network Time Protocol) server, which will be used by the phone to synchronize the date and time.

Allow DHCP Option 42 to override NTP server	DHCP Option 42 specifies a list of IP addresses for Network Time Protocol (NTP) servers available to the client. If you choose yes, BT200 will use the NTP servers resolved from DHCP, instead of the one you specified in the "NTP Server" option above.
Distinctive Ring Tone	Customer Ring Tone 1 to 3 with associate Caller ID: when selected, if Caller ID is configured, then the device will ONLY sound this ring tone when the incoming call is from the Caller ID, device will use System Ring Tone for all other calls. When selected but no Caller ID is configured, the selected ring tone will be used for all incoming calls.
Disable Call Waiting	Default is No.
Quick IP Call Mode	This model has the ability to dial an IP address under the same LAN segment by simply pressing the last octet in the IP address. In the Advanced Settings page there is an option "Use Quick IP-call mode", by default it is set to No. When this option is set to YES, and #XXX is dialed, where X is 0-9 and XXX <=255, phone will make direct IP call to aaa.bbb.ccc.XXX where aaa.bbb.ccc comes from the local IP address REGARDLESS of subnet mask. #XX or #X are also valid so leading 0 is not required (but OK). eg. 192.168.0.2 calling 192.168.0.3 just dial #3 follow by SEND or # 192.168.0.2 calling 192.168.0.23 just dial #23 follow by SEND or # 192.168.0.2 calling 192.168.0.123 just dial #123 follow by SEND or # 192.168.0.2 dial #3 and #03 and #003 has same effect --> call 192.168.0.3 Note:- If you have a SIP Server configured, Direct IP-IP call will still work. However, if you are using STUN, Direct IP-IP call will also use STUN.
Lock keypad update	If this parameter is set to “Yes”, the configuration updates via keypad for Menu Item 7, 9, 12 are disabled.

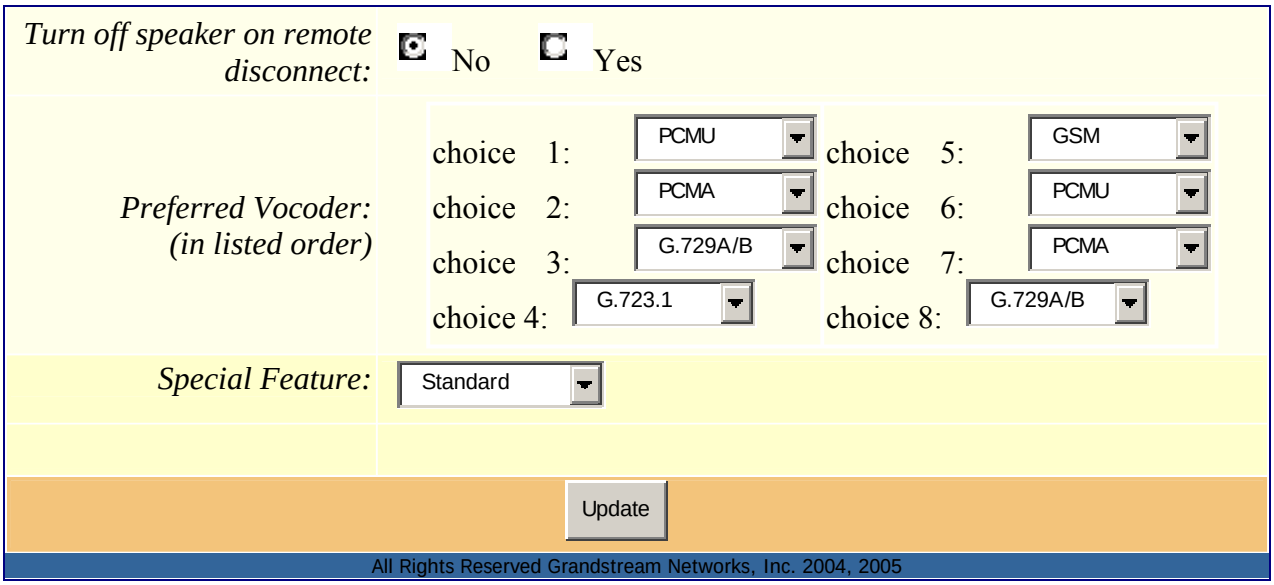
Following is the screenshot of the Account Configuration Page:-

Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS ACCOUNT

Account Active:	○ No ◉ Yes
Account Name:	MyCompany (e.g., MyCompany)
SIP Server:	sip.mycompany.com (e.g., sip.mycompany.com, or IP address)
Outbound Proxy:	(e.g., proxy.myprovider.com, or IP address, if any)
SIP User ID:	123 (the user part of an SIP address)
Authenticate ID:	123 (can be identical to or different from **SIP User ID**)
Authenticate Password:	(purposely not displayed for security protection)
Name:	John Doe (optional, e.g., John Doe)
Use DNS SRV:	◉ No ○ Yes
User ID is phone number:	◉ No ○ Yes
SIP Registration:	○ No ◉ Yes
Unregister On Reboot:	◉ No ○ Yes
Register Expiration:	60 (in minutes. default 1 hour, max 45 days)
local SIP port:	5060 (default 5060)
SIP T1 Timeout:	1 sec
SIP T2 Interval:	4 sec
NAT Traversal (STUN):	○ No ○ No, but send keep-alive ◉ Yes
SUBSCRIBE for MWI:	◉ No ○ Yes
Proxy-Require:	
Voice Mail UserID:	(User ID/extension for 3rd party voice

	mail system)
Send DTMF:	◉ in-audio ○ via RTP (RFC2833) ○ via SIP INFO
Early Dial:	◉ No ○ Yes (use "Yes" only if proxy supports 484 response)
Dial Plan Prefix:	☐ (this prefix string is added to each dialed number)
Enable Call Features:	○ No ◉ Yes (if Yes, Call Forwarding & Call-Waiting-Disable are supported locally)
Session Expiration:	180 (in seconds. default 180 seconds)
Min-SE:	90 (in seconds. default and minimum 90 seconds)
Caller Request Timer:	◉ No ○ Yes (Request for timer when making outbound calls)
Callee Request Timer:	◉ No ○ Yes (When caller supports timer but did not request one)
Force Timer:	◉ No ○ Yes (Use timer even when remote party does not support)
UAC Specify Refresher:	○ UAC ○ UAS ◉ Omit (Recommended)
UAS Specify Refresher:	◉ UAC ○ UAS (When UAC did not specify refresher tag)
Force INVITE:	◉ No ○ Yes (Always refresh with INVITE instead of UPDATE)
Enable 100rel:	◉ No ○ Yes
Account Ring Tone:	◉ system ring tone ○ custom ring tone 1 ○ custom ring tone 2 ○ custom ring tone 3
Send Anonymous:	◉ No ○ Yes (caller ID will be blocked if set to Yes)
Auto Answer:	◉ No ○ Yes
Allow Auto Answer by Call-Info:	◉ No ○ Yes



Individual Account Settings

Account Active	This field indicates whether the account is active or not.
Account Name	A name to identify an account which will be displayed in LCD.
SIP Server	SIP Server's IP address or Domain name provided by VoIP service provider.
Outbound Proxy	IP address or Domain name of Outbound Proxy, or Media Gateway, or Session Border Controller. Used by BudgeTone 200 for firewall or NAT penetration in different network environment. If symmetric NAT is detected, STUN will not work and ONLY outbound proxy can provide solution for it.
SIP User ID	User account information, provided by VoIP service provider (ITSP), usually has the form of digit similar to phone number or actually a phone number.
Authenticate ID	SIP service subscriber's Authenticate ID used for authentication. Can be identical to or different from SIP User ID.
Authenticate Password	SIP service subscriber's account password for BudgeTone 200 to register to (SIP) servers of ITSP.
Name	SIP service subscriber's name which will be used for Caller ID display.

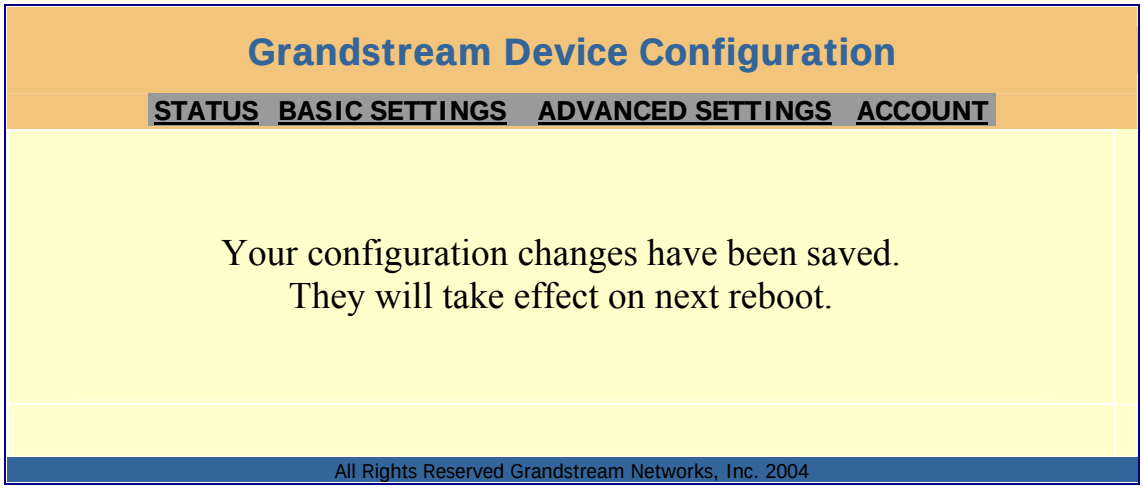
Use DNS SRV:	Default is No. If set to Yes the client will use DNS SRV to look up server.
User ID is Phone Number	If the BudgeTone 200 has an assigned PSTN telephone number, this field should be set to "Yes". Otherwise, set it to "No". If "Yes" is set, a "user=phone" parameter will be attached to the "From" header in SIP request
SIP Registration	This parameter controls whether the BudgeTone 200 needs to send REGISTER messages to the proxy server. The default setting is "Yes".
Unregister on Reboot	Default is No. If set to yes, the SIP user's registration information will be cleared on reboot.
Register Expiration	This parameter allows user to specify the time frequency (in minutes) that BudgeTone 200 refreshes its registration with the specified registrar. The default interval is 60 minutes (or 1 hour). The maximum interval is 65535 minutes (about 45 days).
Local SIP port	This parameter defines the local SIP port the BudgeTone 200 will listen and transmit. The default value for Account 1 is 5060. It is 5062, 5064, 5066 for Account 2, Account 3 and Account 4 respectively.
SIP T1 Timeout	T1 is an estimate of the round-trip time (RTT) between the client and server transactions. If the network latency is high, select bigger value for reliable usage.
SIP T2 Interval	This element sets the value of the SIP protocol T2 timer, in seconds. Timer T2 defines the retransmit interval for INVITE responses and non-INVITE requests. The SIP protocol default value is 4 seconds.
NAT Traversal	This parameter defines whether the BudgeTone 200 NAT traversal mechanism will be activated or not. If activated (by choosing "Yes") and a STUN server is also specified, then the BudgeTone 200 will behave according to the STUN client specification. Under this mode, the embedded STUN client inside the BudgeTone 200 will attempt to detect if and what type of firewall/NAT it is sitting behind through communication with the specified STUN server. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the BudgeTone 200 will attempt to use its mapped public IP address and port in all of its SIP and SDP messages. If the NAT Traversal field is set to "Yes" with no specified STUN server, the BudgeTone 200 will periodically (every 20 seconds or so) send a blank UDP packet (with no payload data) to the SIP server to keep the "hole" on the NAT open.
Subscribe for MWI:	Default is No. When set to "Yes" a SUBSCRIBE for Message Waiting Indication will be sent periodically.

Proxy-Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
Voice Mail User ID	When configured, user will be able to dial voice mail server by pressing "MSG" button.
Send DTMF	This parameter specifies the mechanism to transmit DTMF digit. There are 3 modes supported: in audio which means DTMF is combined in audio signal (not very reliable with low-bit-rate codec), via RTP (RFC2833), or via SIP INFO.
Early Dial	Default is No. Use only if proxy supports 484 response.
Dial Plan Prefix	Sets the prefix added to each dialed number.
Enable Call Features	Default is No. If set to Yes, Call transfer, Call Forwarding & Do-Not-Disturb are supported locally.
Disable Missed-Call	Default is No. If set to Yes, missed calls will not be recorded for your review.
Session Expiration	Grandstream implemented SIP Session Timer. The session timer extension enables SIP sessions to be periodically "refreshed" via a SIP request (UPDATE, or re-INVITE. Once the session interval expires, if there is no refresh via a UPDATE or re-INVITE message, the session will be terminated. Session Expiration is the time (in seconds) at which the session is considered timed out, if no successful session refresh transaction occurs beforehand. The default value is 180 seconds.
Min-SE	The minimum session expiration (in seconds). The default value is 90 seconds.
Caller Request Timer	If selecting "Yes" the phone will use session timer when it makes outbound calls if remote party supports session timer.
Callee Request Timer	If selecting "Yes" the phone will use session timer when it receives inbound calls with session timer request.
Force Timer	If selecting "Yes" the phone will use session timer even if the remote party does not support this feature. Selecting "No" will allow the phone to enable session timer only when the remote party support this feature. To turn off Session Timer, select "No" for Caller Request Timer, Callee Request Timer, and Force Timer.
UAC Specify Refresher	As a Caller, select UAC to use the phone as the refresher, or UAS to use the Callee or proxy server as the refresher.

UAS Specify Refresher	As a Callee, select UAC to use caller or proxy server as the refresher, or UAS to use the phone as the refresher.
Force INVITE	Session Timer can be refreshed using INVITE method or UPDATE method. Select "Yes" to use INVITE method to refresh the session timer.
Enable 100rel	The use of the PRACK (Provisional Acknowledgment) method enables reliability to be offered to SIP provisional responses (1xx series). This is very important if PSTN internetworking is to be supported. A user's wish to use reliable provisional responses is invoked by the 100rel tag which is appended to the value of the required header of initial signalling messages.
Account Ring Tone	There are 4 different ring tone that are defined: • System Ring Tone: when selected, all calls will ring with system ring tone. • Customer Ring Tone 1 to 3: when selected, BudgeTone 200 will ONLY play this ring tone for all the incoming calls for this account.
Send Anonymous	If this parameter is set to "Yes", the "From" header in outgoing INVITE message will be set to anonymous, essentially blocking the Caller ID from displaying.
Auto Answer	When set to "Yes", BudgeTone 200 will automatically switch to speaker when there is an incoming call.
Allow Auto Answer by Call-Info	Default is No. If set to Yes, auto answer depends on the Call-Info in the SIP message. This feature needs the support of IP-PBX.
Turn off speaker on remote disconnect	Default is No. If set to Yes, the speaker will turn off, and the phone will go back to idle status, after the other party of the call hands up.
Preferred Vocoder	The BudgeTone 200 supports up to 5 different Vocoder types including G.711 A-/U-law, GSM, G.723.1, G.729A/B. User can configure Vocoders in a preference list that will be included with the same preference order in SDP message. The first Vocoder in this list can be entered by choosing the appropriate option in "Choice 1". Similarly, the last Vocoder in this list can be entered by choosing the appropriate option in "Choice 8".
Special Feature	Default is Standard. Choose the selection to meet some special requirements from Soft Switch vendors like Nortel, Broadsoft, etc.

5.2.4 Saving the Configuration Changes

Once a change is made, the user should press the "Update" button in the Configuration Menu. The IP phone will then display the following screen to confirm that the changes have been saved:


Grandstream Device Configuration
STATUS BASIC SETTINGS ADVANCED SETTINGS ACCOUNT
Your configuration changes have been saved.
They will take effect on next reboot.
All Rights Reserved Grandstream Networks, Inc. 2004

User is recommended to power cycle the IP phone after seeing the above message.

5.2.5 Rebooting the Phone from Remote

The administrator of the phone can remotely reboot the phone by pressing the "Reboot" button at the bottom of the configuration menu. Once done, the following screen will be displayed to indicate that rebooting is underway.

Grandstream Device Configuration
The device is rebooting now...
You may relogin by clicking on the link below in 30 seconds.
Click to relogin
All Rights Reserved Grandstream Networks, Inc. 2004

At this point, user can relogin to the phone after waiting for about 30 seconds.

5.3 Configuration through Central Provisioning Server

Grandstream BudgeTone 200 can be automatically configured from a central provisioning system.

When BudgeTone 200 boots up, it will send TFTP or HTTP request to download configuration files, there are two configuration files, one is "cfg.txt" and the other is "cfg000b82xxxxxx", where "000b82xxxxxx" is the MAC address of the BudgeTone 200.

The configuration files can be downloaded via TFTP or HTTP from the central server. A service provider or an enterprise with large deployment of BudgeTone 200 can easily manage the configuration and service provisioning of individual devices remotely from a central server.

Grandstream provides a licensed provisioning system called GAPS that can be used to support automated configuration of BudgeTone 200. GAPS (Grandstream Automated Provisioning System) uses enhanced (NAT friendly) TFTP or HTTP (thus no NAT issues) and other communication protocols to communicate with each individual BudgeTone 200 for firmware upgrade, remote reboot, etc.

Grandstream provide GAPS (Grandstream Automated Provisioning System) service to VoIP service providers. It could be either simple redirection or with certain special provisioning settings. Initially upon booting up, Grandstream devices by default point to Grandstream provisioning server GAPS, based on the unique MAC address of each device, GAPS provision the devices with redirection settings so that they will be redirected to customer's TFTP or http server for further provisioning.
Grandstream also provide GAPSLite software package which contains our NAT friendly TFTP server and a configuration tool to facilitate the task of generating device configuration files.

The GAPSLite configuration tool is now free to end users. The tool and configuration templates can be downloaded from
http://www.grandstream.com/DOWNLOAD/Configuration_Tool/.

For details on how GAPS works, please refer to the documentation of GAPS product.

6 Firmware Upgrade

6.1 Upgrade through HTTP

To upgrade software, BudgeTone 200 can be configured with an HTTP server where the new code image file is located. For example, following URL in the HTTP Upgrade Server:

http://firmware.mycompany.com:6688/Grandstream/1.0.1.12

Where firmware.mycompany.com is the FQDN of the HTTP server, ":6688" is the TCP port the HTTP server listening to, "/Grandstream/1.0.0.4" is the RELATIVE directory to the root dir in HTTP server. Thus, you can put different firmware into different directory as well.

NOTE:

- *If "Auto Upgrade" field is set to "No", HTTP upgrade will be performed only once during boot up. If it is set to "Yes", the device will check the HTTP server in the number of days that is defined in "Check for new firmware every" field.*

6.2 Upgrade through TFTP

To upgrade software, BudgeTone 200 can be configured with a TFTP server where the new code image is located. It is recommended to set the TFTP server address in either a public IP address or on the same LAN with the BudgeTone 200.

There are two ways to set up the TFTP server to upgrade the firmware, namely through voice menu prompt or via the BudgeTone 200's Web configuration interface. To configure the TFTP server via voice prompt, please refer to section 5.1 with option 06, once set up the TFTP IP address, power cycle the device, the firmware will be fetched once the device boots up.

To configure the TFTP server via the Web configuration interface, open up your browser to point at the IP address of the BudgeTone 200. Input the admin password to enter the configuration screen. From there, enter the TFTP server address in the designated field towards the bottom of the configuration screen. Once the TFTP server is set, user needs to update the change by clicking the "Update" button. Then "Reboot" or power cycle the phone, the firmware files will be fetched upon booting up.

TFTP checking is only performed during the initial power up. If the configured TFTP server is found and a new code image is available, the BudgeTone 200 will attempt to

retrieve the new image files by downloading them into the BudgeTone 200's SRAM. During this stage, the BudgeTone 200's LEDs will blink until the checking/downloading process is completed. Upon verification of checksum, the new code image will then be saved into the Flash. If TFTP fails for any reason (e.g., TFTP server is not responding, there are no code image files available for upgrade, or checksum test fails, etc), the BudgeTone 200 will stop the TFTP process and simply boot using the existing code image in the flash.

TFTP process may take as long as 1 to 2 minutes over the Internet, or just 20+ seconds if it is performed on a LAN. Users are recommended to conduct TFTP upgrade in a controlled LAN environment if possible. For those who do not have a local TFTP server, Grandstream provides a NAT-friendly TFTP server on the public Internet for firmware upgrade. Please check the Services section of Grandstream's Web site to obtain this TFTP server's IP address.

NOTE:

- *When BudgeTone 200 boots up, it will send TFTP or HTTP request to download configuration files, there are two configuration files, one is "cfg.txt" and the other is "cfg000b82xxxxxx", where "000b82xxxxxx" is the MAC address of the BudgeTone 200. These two files are for initial automatically provisioning purpose only, for normal TFTP or HTTP firmware upgrade, the following error messages in a TFTP or HTTP server log can be ignored.*

```
   TFTP Error from [IP ADRESS] requesting cfg000b82023dd4 : File
does not exist
   TFTP Error from [IP ADRESS] requesting cfg.txt : File does not
exist
```

7 Restore Factory Default Setting

Warning !!!

Restore the Factory Default Setting will DELETE all configuration information of the device. Please backup or print out all the settings before you approach to following steps. Grandstream will not take any responsibility if you lose all the parameters of setting and cannot connect to your service provider.

Please disconnect network cable and power cycle the unit before trying to reset the unit to factory default. The steps are as follows:

Step 1:
Find the MAC address of the device. It is a 12 digits HEX number located on the bottom of the unit.

Step 2:
Encode the MAC address. Please use the following mapping:

0-9: 0-9
A: 22 (when pressed 2 twice, the "A" letter will show on the LCD)
B: 222
C: 2222
D: 33
E: 333
F: 3333

For example, if the MAC address is 000b8200e395, it should be encoded as "0002228200333395".

Step 3:
To perform factory reset:

a. Press the **MENU** button for Key Pad Menu options.
b. Press the Up or Down button to see "reset".
c. Enter the encoded MAC address.
d. Press the **MENU** button again
e. Wait for phone reboot and the LCD backlight finish flashing.

8 Appendix I Glossary of Terms

ADSL

Asymmetric Digital Subscriber Line: Modems attached to twisted pair copper wiring that transmit from 1.5 Mbps to 9 Mbps downstream (to the subscriber) and from 16 kbps to 800 kbps upstream, depending on line distance.

AGC

Automatic Gain Control, is an electronic system found in many types of devices. Its purpose is to control the gain of a system in order to maintain some measure of performance over a changing range of real world conditions.

ARP

Address Resolution Protocol is a protocol used by the Internet Protocol (IP) [RFC826], pecifically IPv4, to map IP network addresses to the hardware addresses used by a data link protocol. The protocol operates below the network layer as a part of the interface between the OSI network and OSI link layer. It is used when IPv4 is used over Ethernet

ATA

Analogue Telephone Adapter. Covert analogue telephone to be used in data network for VoIP, like Grandstream HT series products.

CODEC

Abbreviation for Coder-Decoder. It's an analog-to-digital (A/D) and digital-to-analog (D/A) converter for translating the signals from the outside world to digital, and back again.

CNG

Comfort Noise Generator, geneate artificial background noise used in radio and wireless communications to fill the silent time in a transmission resulting from voice activity detection.

DATAGRAM

A data packet carrying its own address information so it can be independently routed from its source to the destination computer

DECIMATE

To discard portions of a signal in order to reduce the amount of information to be encoded or compressed. Lossy compression algorithms ordinarily decimate while subsampling.

DECT

Digital Enhanced Cordless Telecommunications: A standard developed by the European Telecommunication Standard Institute from 1988, governing pan-

European digital mobile telephony. DECT covers wireless PBXs, telepoint, residential cordless telephones, wireless access to the public switched telephone network, Closed User Groups (CUGs), Local Area Networks, and wireless local loop. The DECT Common Interface radio standard is a multicarrier time division multiple access, time division duplex (MC-TDMA-TDD) radio transmission technique using ten radio frequency channels from 1880 to 1930 MHz, each divided into 24 time slots of 10ms, and twelve full-duplex accesses per carrier, for a total of 120 possible combinations. A DECT base station (an RFP, Radio Fixed Part) can transmit all 12 possible accesses (time slots) simultaneously by using different frequencies or using only one frequency. All signaling information is transmitted from the RFP within a multiframe (16 frames). Voice signals are digitally encoded into a 32 kbit/s signal using Adaptive Differential Pulse Code Modulation.

DNS

Short for ***D****omain* ***N****ame* ***S****ystem* (or ***S****ervice* or ***S****erver*), an Internet service that translates *domain names* into IP addresses

DID

Direct Inward Dialing

Direct Inward Dialing. The ability for an outside caller to dial to a PBX extension without going through an attendant or auto-attendant.

DSP

Digital Signal Processing. Using computers to process signals such as sound, video, and other analog signals which have been converted to digital form.

Digital Signal Processor. A specialized CPU used for digital signal processing.

Grandstream products all have DSP chips built inside.

DTMF

Dual Tone Multi Frequency

The standard tone-pairs used on telephone terminals for dialing using in-band signaling. The standards define 16 tone-pairs (0-9, #, * and A-F) although most terminals support only 12 of them (0-9, * and #).

FQDN

Fully Qualified Domain Name

A FQDN consists of a host and domain name, including top-level domain. For example, www.grandstream.com is a fully qualified domain name. www is the host, grandstream is the second-level domain, and.com is the top level domain.

FXO

Foreign eXchange Office

An FXO device can be an analog phone, answering machine, fax, or anything that handles a call from the telephone company like AT&T. They should also operate the same way when connected to an FXS interface.

An FXO interface will accept calls from FXS or PSTN interfaces. All countries and regions have their own standards.

FXO is complimentary to FXS (and the PSTN).

FXS

Foreign eXchange Station

An FXS device has hardware to generate the ring signal to the FXO extension (usually an analog phone).

An FXS device will allow any FXO device to operate as if it were connected to the phone company. This makes your PBX the POTS+PSTN for the phone.

The FXS Interface connects to FXO devices (by an FXO interface, of course).

DHCP

The *Dynamic Host Configuration Protocol* (DHCP) is an Internet protocol for automating the configuration of computers that use TCP/IP. DHCP can be used to automatically assign IP addresses, to deliver TCP/IP stack configuration parameters such as the subnet mask and default router, and to provide other configuration information such as the addresses for printer, time and news servers.

ECHO CANCELLATION

Echo Cancellation is used in telephony to describe the process of removing echo from a voice communication in order to improve voice quality on a telephone call. In addition to improving quality, this process improves bandwidth savings achieved through silence suppression by preventing echo from traveling across a network.

There are two types of echo of relevance in telephony: acoustic echo and hybrid echo. Speech compression techniques and digital processing delay often contribute to echo generation in telephone networks.

H.323

A suite of standards for multimedia conferences on traditional packet-switched networks.

HTTP

Hyper Text Transfer Protocol; the World Wide Web protocol that performs the request and retrieve functions of a server

IP

Internet Protocol. A packet-based protocol for delivering data across networks**.**

IP-PBX

IP-based Private Branch Exchange

IP Telephony

(Internet Protocol telephony, also known as Voice over IP Telephony) A general term for the technologies that use the Internet Protocol's packet-switched connections to exchange voice, fax, and other forms of information that have traditionally been carried over the dedicated circuit-switched connections of the public switched telephone network (PSTN). The basic steps involved in originating an IP Telephony call are conversion of the analog voice signal to digital format and compression/translation of the signal into Internet protocol (IP) packets for transmission over the Internet or other packet-switched networks; the process is reversed at the receiving end. The terms IP Telephony and Internet Telephony are often used to mean the same; however, they are not 100 per cent interchangeable, since Internet is only a subcase of packet-switched networks. For users who have free or fixed-price Internet access, IP Telephony software essentially provides free telephone calls anywhere in the world. However, the challenge of IP Telephony is maintaining the quality of service expected by subscribers. Session border controllers resolve this issue by providing quality assurance comparable to legacy telephone systems.

IVR

IVR is a software application that accepts a combination of voice telephone input and touch-tone keypad selection and provides appropriate responses in the form of voice, fax, callback, e-mail and perhaps other media.

MTU

A Maximum Transmission Unit (MTU) is the largest size packet or frame, specified in octets (eight-bit bytes), that can be sent in a packet- or frame-based network such as the Internet. The maximum for Ethernet is 1500 byte.

NAT

Network Address Translation

NTP

Network Time Protocol, a protocol to exchange and synchronize time over networks
The port used is UDP 123
Grandstream products using NTP to get time from Internet

OBP/SBC

Outbound Proxy or another name Session Border Controller. A device used in VoIP networks. OBP/SBCs are put into the signaling and media path between calling and called party. The OBP/SBC acts as if it was the called VoIP phone and places a second call to the called party. The effect of this behaviour is that not only the signaling traffic, but also the media traffic (voice, video etc) crosses the OBP/SBC. Without an OBP/SBC, the media traffic travels directly between the VoIP phones. Private OBP/SBCs are used along with firewalls to enable VoIP calls to and from a protected enterprise network. Public VoIP service providers use OBP/SBCs to allow the use of VoIP protocols from private networks with internet connections using NAT.

PPPoE

Point-to-Point Protocol over Ethernet, is a network protocol for encapsulating PPP frames in Ethernet frames. It is used mainly with cable modem and DSL services.

PSTN

Public Switched Telephone Network

i.e. the phone service we use for every ordinary phone call, or called POT (Plain Old Telephone), or circuit switched network.

RTCP

Real-time Transport Control Protocol, defined in RFC 3550, a sister protocol of the Real-time Transport Protocol (RTP), It partners RTP in the delivery and packaging of multimedia data, but does not transport any data itself. It is used periodically to transmit control packets to participants in a streaming multimedia session. The primary function of RTCP is to provide feedback on the quality of service being provided by RTP.

RTP

Real-time Transport Protocol defines a standardized packet format for delivering audio and video over the Internet. It was developed by the Audio-Video Transport Working Group of the IETF and first published in 1996 as RFC 1889

SDP

Session Description Protocol, is a format for describing streaming media initialization parameters. It has been published by the IETF as RFC 2327.

SIP

Session Initiation Protocol, An IP telephony signaling protocol developed by the IETF (RFC3261). SIP is a text-based protocol suitable for integrated voice-data applications. SIP is designed for voice transmission and uses fewer resources and is considerably less complex than H.323.
All Grandstream products are SIP based

STUN

Simple Traversal of UDP over NATs, is a network protocol allowing clients behind NAT (or multiple NATs) to find out its public address, the type of NAT it is behind and the internet side port associated by the NAT with a particular local port. This information is used to set up UDP communication between two hosts that are both behind NAT routers. The protocol is defined in RFC 3489. STUN will usually work good with non-symmetric NAT routers.

TCP

Transmission Control Protocol, is one of the core protocols of the Internet protocol suite. Using TCP, applications on networked hosts can create connections to one another, over which they can exchange data or packets. The protocol guarantees reliable and in-order delivery of sender to receiver data.

TFTP

Trivial File Transfer Protocol, is a very simple file transfer protocol, with the functionality of a very basic form of FTP; It uses UDP (port 69) as its transport protocol.

UDP

User Datagram Protocol (UDP) is one of the core protocols of the Internet protocol suite. Using UDP, programs on networked computers can send short messages known as datagrams to one another. UDP does not provide the reliability and ordering guarantees that TCP does; datagrams may arrive out of order or go missing without notice. However, as a result, UDP is faster and more efficient for many lightweight or time-sensitive purposes.

VAD

Voice Activity Detection or Voice Activity Detector is an algorithm used in speech processing wherein, the presence or absence of human speech is detected from the audio samples.

VLAN

A virtual LAN, known as a VLAN, is a logically-independent network. Several VLANs can co-exist on a single physical switch. It is usually refer to the IEEE 802.1Q tagging protocol.

VoIP

Voice over IP

VoIP encompasses many protocols. All the protocols do some form of signalling of call capabilities and transport of voice data from one point to another. e.g: SIP, H.323, etc.